

**APPLICATION FOR
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FOR

Method and Apparatus for Receiving Radio Frequency Signals

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Method and Apparatus for Receiving Radio Frequency Signals

CROSS-REFERENCE TO RELATED APPLICATION

5 This application is a continuation-in-part of PCT International Application No. PCT/US00/12298, entitled "Receiver Circuit", filed on May 4th, 2000, published under PCT Article 21(2) in English, which PCT application claims priority to Great Britain Application Number 9910662.7, entitled "Pass-Band Signal Processing" filed on May 7th, 1999. This application therefore claims priority under 35 U.S.C. §§ 120 and 363 to the PCT Application
10 No.: PCT/US00/12298 filed May 4th, 2000.

BACKGROUND

1. *Field*

15 This invention relates to receiving method and apparatus, and more particularly to a method and apparatus for receiving radio frequency signals.

2. *Description of Related Art*

20 Conventional radio receiver circuits in which a received analog signal is downconverted in a first mixer stage to a first intermediate frequency, and subsequently downconverted in a second mixer stage to a second intermediate frequency are well known. It is also well known to sample the analog signal at the second intermediate frequency using an analog-to-digital converter.

25 Also well known are techniques of digital sub-sampling, whereby an analog-to-digital converter is used to achieve downconversion of a signal. These techniques rely upon the well-known phenomenon of signal "aliasing". An analog-to-digital converter having a sampling rate (or sample frequency) of F can only entirely reliably reproduce signals having a frequency
30 below (*i.e.*, less than) $F/2$. Higher frequency signals are still detected, but these signals appear in the output digital signal at frequencies ranging from 0 to $F/2$. Thus, analog input signals

having frequencies of f , $(F-f)$, $(F+f)$, $(2F-f)$, $(2F+f)$, *etc.* appear in an output signal at the frequency f .

5 The prior art digital sub-sampling techniques are utilized in a well-known manner to achieve downconversion of radio frequency signals. For example, one such prior art system is taught by Bella *et al.*, in U.S. Patent No. 5,630,227, issued on May 13, 1997. In particular, digital sub-sampling techniques can be used to downconvert a signal that only has components over a relatively narrow range of frequencies. For example, if an analog signal has frequency components only at one or more frequencies (designated $(3F+f)$) within a range from $3F$ to $3.5F$, and is sampled by an analog digital converter at a sampling frequency F , the output
10 digital signal will have corresponding components at the frequency or frequencies f in the range from 0 to $0.5F$. In other words, the frequency range from $3F$ to $3.5F$ is said to be "aliased" to a range from 0 to $0.5F$.

15 Disadvantageously, the above-described well-known system is unable to effectively combat the detrimental effects of adjacent channel interference. Specifically, when a signal has a frequency that is relatively close to a frequency of one of the desired signals in the input, the above-described known system causes this signal to produce an output that interferes with the desired output signals in an unpredictable manner. In other words, the interferer (*i.e.*, the
20 signal that has a frequency that is relatively close to a frequency of one of the desired input signals) may alias to a frequency close to that at which a desired output signal will appear, and moreover may be a stronger signal than the desired signal, such that it cannot easily be removed through filtering.

25 Therefore, a need exists for a method and apparatus for receiving radio frequency signals in a communication system that can be easily implemented and overcomes the disadvantages of other methods and apparatuses such as the above-described known systems. The present disclosure provides such a radio frequency receiver method and apparatus.

SUMMARY

5 This disclosure describes a method and apparatus for receiving radio frequency signals. The present method and apparatus counteracts aliasing problems associated with the prior art techniques by determining a relationship between the center frequency of an analog-to-digital converter input signal, a frequency of an undesired or unwanted signal, and the sampling rate of the analog-to-digital converter.

10 In one embodiment a radio receiver circuit is described, wherein the radio receiver circuit receives an input signal in a received signal band, including a desired or wanted signal in a first desired, or wanted frequency band between a lower wanted frequency and an upper wanted frequency, the wanted signal being centered at a first wanted frequency band center frequency, and the input signal further including an interference signal at an interference frequency within the received signal band. The receiver circuit comprises an analog-to-digital
15 converter having a sampling frequency that is less than twice the upper wanted frequency for downconverting the input signal. The sampling frequency is selected such that the degree of aliasing of the interference signal into the first wanted frequency band after downconversion is kept below a predetermined threshold.

20 In a second embodiment, a method of receiving an input radio signal in a received signal band is described. In this embodiment, the input signal includes a desired or a wanted signal in a first wanted frequency band between a lower wanted frequency and an upper wanted frequency. The desired or wanted signal is centered at a first wanted frequency band center frequency, and further includes an interference signal at an interference frequency within the
25 received signal band. The method includes the step of downconverting the input signal by sampling the input signal at a sampling frequency that is less than twice the upper wanted frequency. The sampling frequency is selected such that the degree of aliasing of the interference signal into the first wanted frequency band after downconversion is maintained below a predetermined threshold.

30 Thus, the sampling frequency is chosen relative to the first wanted frequency band center frequency, which advantageously is a first intermediate frequency after initial downconversion

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of the input signal, such that the interference signal is maintained after subsampling, allowing the interference signal to be removed at baseband.

BRIEF DESCRIPTION OF THE DRAWINGS

FIGURE 1 is a simplified block diagram of a receiver circuit in accordance with one embodiment of the invention.

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FIGURE 2 is a first exemplary graphic representation showing the aliasing of received signals in the embodiment shown in FIGURE 1.

FIGURE 3 is a second exemplary graphic representation showing the aliasing of received signals in the embodiment shown in FIGURE 1.

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FIGURE 4 is a third exemplary graphic representation showing the aliasing of received signals in the embodiment shown in FIGURE 1.

FIGURE 5 is a fourth exemplary graphic representation showing the aliasing of received signals in the embodiment shown in FIGURE 1.

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FIGURE 6 is a fifth exemplary graphic representation showing the aliasing of received signals in the embodiment shown in FIGURE 1.

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Like reference numbers and designations in the various drawings indicate like elements.

DETAILED DESCRIPTION OF THE INVENTION

Throughout this description, the preferred embodiment and examples shown should be considered as exemplars, rather than as limitations to the present invention.

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The disclosed devices and methods are methods and apparatus for receiving radio frequency signals in communication systems. The present inventive method and apparatus utilizes an inventive receiver circuit that downconverts input signals so that interference components of the input signals can be easily removed at baseband. The present inventive method and apparatus is now described in detail with reference to FIGURE 1.

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FIGURE 1 shows a receiver circuit made in accordance with the present invention. The invention is described herein with reference to its application in the reception of digital terrestrial television (DTT) signals using the European DVB-T standard based on Coded Orthogonal Frequency Division Multiplexing (COFDM). However, those skilled in the receiver arts shall appreciate that the present invention's use is independent of the type of signals being received. The present invention can be used to receive signals in virtually any type of radio frequency communication system.

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FIGURE 1 shows an antenna 2, for receiving broadcast UHF/VHF signals, containing video data modulated using Coded Orthogonal Frequency Division Multiplexing (COFDM). The broadcast signals are supplied to an analog tuner 4 as shown in FIGURE 1. The tuner 4 includes a mixer 6 which receives a first local oscillator signal LO1 used for the downconversion of the received signals to a first intermediate frequency, and a band-pass filter 8, which may, for example, be formed from a pair of SAW filters. The filter 8 is assumed to attenuate all signals outside of a channel of width CW, at least to a level at which they cannot interfere with desired or wanted received signals. A conventional downconversion process will typically invert the frequency sense of the received signal spectrum.

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The output signal that is produced by the analog tuner 4, at an output 10, is therefore at a first intermediate frequency "IF1". The IF1 signal is applied to an automatic gain control circuit 11, and then to an analog-to-digital converter 12. The analog-to-digital converter has a

sampling rate SR which is less than twice the first intermediate frequency IF1, and therefore sub-samples the signal. This sub-sampling effectively downconverts the signal by aliasing to a second intermediate frequency, "IF2" which is relatively close to baseband. The analog-to-digital converter 12 should therefore be designed to have an adequate response to signals at the first intermediate frequency IF1. The automatic gain control circuit 11 is capable of maintaining the signal level of the first intermediate frequency IF1 so that the analog-digital converter 12 can accurately sample the IF1 signal.

The baseband output from the analog-to-digital converter 12 is supplied to a filtering device 13, and then to a demodulator 14 in the form of digitized samples of the input signals. The filtering device 13 includes a mixer 16, which receives a second local oscillator signal "LO2". The second local oscillator signal LO2 is at the second intermediate frequency "IF2". The output from the mixer 16 is input to a low-pass filter 18, for removal of undesired or unwanted components. The demodulator 14 removes the COFDM modulation, and supplies output signals that can be converted into a form that is suitable for display.

Although FIGURE 1 shows several discrete blocks, it will be appreciated by those skilled in the electronics design arts that the different stages may be integrated as much as is desirable, for example onto a single chip, or other arrangements of functions can be used. For example, the analog tuner 4 may comprise one component, while the analog-to-digital converter 12, demodulator 14, and subsequent processing circuitry may be combined.

FIGURE 2 shows a first exemplary graphic representation of a signal present at the output 10 of the tuner 4. As shown in FIGURE 2, the downconverted signal is centered at the first intermediate frequency IF1, as described above. The band-pass filter 8 has a channel width CW centered at IF1, thus signals in the range $(IF1 - CW * \frac{1}{2})$ to $(IF1 + CW * \frac{1}{2})$ appear at the output 10. The shaded area 20 represents the signal bandwidth SB, which contains desired or "wanted" COFDM signals.

However, because the channel width CW is sufficiently wide to pass not only the desired signal bandwidth SB, it is also sufficiently wide to pass any adjacent, potentially interfering undesired signal. For example, the unwanted signal may appear at $(IF1 + N)$. For example, in the United Kingdom, a NICAM (Near Instantaneous Companding Audio Multiplex) sound

signal may appear at this point. Moreover, the NICAM signal may be strong (for example +10dB) relative to the desired or wanted COFDM signals.

5 In principle, it would be possible to design the band-pass filter 8 such that this unwanted signal is filtered out at that point. However, the gap between the edge of the wanted signal bandwidth and the adjacent unwanted signal is relatively narrow, at least compared to the intermediate frequency IF1, and so it is relatively difficult to achieve this filtering at the intermediate frequency. It is preferable to be able to filter out this unwanted signal at baseband, but, in order to be able to do this, it is necessary to avoid a situation where the
10 unwanted signal appears within the wanted signal in the downconverted signal as a result of aliasing.

The present invention relates to a method and apparatus that removes unwanted signals from wanted signals in the downconverted signal. Consequently, a tuner can be designed having a
15 single downconversion stage, without placing excessive demands on the filter or filters in the tuner. Moreover, one aspect of the present invention involves maintaining the interfering signal unaffected, right until it is removed. Thus, the analog-to-digital converter 12 must have sufficient headroom, that is, enough effective bits, to be able to accurately represent both the interfering signal and the wanted signal. Further, the automatic gain control circuit 11 scales
20 the tuner output so that it fits optimally into the available range of the analog-to-digital converter.

FIGURE 3 shows a possible situation after sub-sampling, at the output of the analog-to-digital converter 12. In this case, the sampling rate SR has been chosen such that the whole of the
25 tuner pass-band from $(IF1 - CW * \frac{1}{2})$ to $(IF1 + CW * \frac{1}{2})$ appears within the frequency range from $(k - \frac{1}{2}) * SR$ to $k * SR$, where k is an integer. After sub-sampling, the entire tuner pass-band appears, inverted, in the frequency range from 0 to $\frac{1}{2} * SR$. In particular, if the center frequency of the pass-band, the intermediate frequency IF1, is separated from the relevant multiple of the sampling frequency $k * SR$ by a frequency separation FS1, where FS1 equals
30 $((k * SR) - IF1)$, then the center frequency of the downconverted signal appears at FS1, which is, in effect, a second intermediate frequency at close to baseband.

If the first downconversion stage inverts the frequency sense of the spectrum, this re-inversion is desirable. However, this inversion can later be removed if necessary, by inverting the sign of all Q values in the I and Q digital samples.

5 As shown in FIGURE 3, the pass-band from $(IF1 - CW^{*1/2})$ to $(IF1 + CW^{*1/2})$ aliases to the range from $(FS1 - CW^{*1/2})$ to $(FS1 + CW^{*1/2})$, while the potentially interfering unwanted signal aliases from $(IF1 + N)$ to $(FS1 - N)$. Because the unwanted signal remains outside the signal band SB, which is now centered on FS1, it can relatively easily be filtered out in the demodulator 14 before the signal is processed further. Specifically, the signal is preferably mixed in a mixer
10 16 with a complex carrier at FS1. The unwanted signal, which is further removed from FS1 than is the wanted signal, is mixed to a higher frequency, and can be removed by a low-pass filter 18, to an extent sufficient to avoid affecting further processes. If necessary, a second automatic gain control circuit (not shown) can be used to boost the signal to an appropriate level.

15 FIGURE 4 shows an alternative possible situation after sub-sampling, at the output of the analog-to-digital converter 12. In this case, the sampling rate SR has been chosen such that $k*SR$, where k is an integer, falls within the tuner pass-band from $(IF1 - CW^{*1/2})$ to $(IF1 + CW^{*1/2})$. After sub-sampling, that part of the tuner pass-band from $(IF1 - CW^{*1/2})$ to $k*SR$ appears, inverted, in the frequency range from 0 to $1/2*SR$. Further, however, that part of the
20 tuner pass-band from $k*SR$ to $(IF1 + CW^{*1/2})$ also appears, uninverted, in the frequency range from 0 to $1/2*SR$.

In effect, the aliasing means that the upper end of the tuner pass-band seems to reflect about
25 the zero frequency point in the downconverted signal. In this case, if the center frequency of the pass-band, the intermediate frequency IF1, is separated from the relevant multiple of the sampling frequency $k*SR$ by a frequency separation FS2 (where FS2 equals $((k*SR) - IF1)$) then the center frequency of the downconverted signal appears at FS2.

30 As described above, the part of the pass-band from $(IF1 - CW^{*1/2})$ to $k*SR$ aliases to the range from 0 to $(FS2 + CW^{*1/2})$, while the part of the pass-band from $k*SR$ to $(IF1 + CW^{*1/2})$ aliases from 0 to $(IF1 + CW^{*1/2} - k*SR)$, or, said in other words, from 0 to $(1/2*CW - FS2)$. There should be no aliasing of the COFDM wanted signal into itself. That is, in FIGURE 4, the upper end of the wanted signal, at $(IF1 + SB^{*1/2})$ aliases to $(FS2 - SB^{*1/2})$, and it is therefore necessary that

$(FS2 - SB * \frac{1}{2}) > 0$. Further, and in particular, the potentially interfering unwanted signal at $(IF1 + N)$ aliases to $(FS2 - N)$, if $FS2 > N$. The potentially interfering unwanted signal at $(IF1 + N)$ aliases to $(N - FS2)$, if $N > FS2$.

5 In order to allow the unwanted signal to be filtered out in the demodulator 14, it should remain outside of the signal band SB, which is now centered on FS2. Moreover, the unwanted signal should be sufficiently far outside the signal band to be filtered therefrom, even allowing for any frequency offset that may be present.

10 If $FS2 > N$, then because $N > SB * \frac{1}{2}$ (because the unwanted signal is known to appear outside the wanted signal band in the signal at the first intermediate frequency), the unwanted signal will be aliased outside of the wanted signal band. However, if $N > FS2$, it is possible that the unwanted signal will be aliased into the wanted signal band. In order to avoid this, it is therefore desirable that the method adheres to the following condition:

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$$(N - FS2) + \Delta < (FS2 - SB * \frac{1}{2});$$
where Δ is the allowed frequency offset, or reflects the fact that the unwanted signal at $(IF1 + N)$ may have a finite bandwidth and be centered at that frequency.

20 Conversely, if the sampling rate SR is chosen such that $(k - \frac{1}{2}) * SR$ falls within the pass-band, the part of the pass-band from $(IF1 - CW * \frac{1}{2})$ to $(k - \frac{1}{2}) * SR$ aliases to the range from 0 to $\frac{1}{2} * SR$ without frequency inversion, while the part of the pass-band from $(k - \frac{1}{2}) * SR$ to $(IF1 + CW * \frac{1}{2})$ also aliases into the range from 0 to $\frac{1}{2} * SR$, with frequency inversion. In effect, this aliasing means that the lower end of the tuner pass-band seems to reflect about the $\frac{1}{2} * SR$ frequency point in the downconverted signal.

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FIGURE 5 shows this reflection about the $\frac{1}{2} * SR$ frequency point in the downconverted signal. In this case, if the center frequency of the pass-band, the intermediate frequency IF1, is separated from the relevant multiple of the sampling frequency $k * SR$ by a frequency separation FS3, where FS3 equals $(k * SR - IF1)$, then the center frequency of the downconverted signal appears at FS3.

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As mentioned above, the part of the pass-band from $(IF1 - CW * \frac{1}{2})$ to $(k - \frac{1}{2}) * SR$ aliases to the range from 0 to $\frac{1}{2} * SR$, while the part of the pass-band from $(k - \frac{1}{2}) * SR$ to $(IF1 + CW * \frac{1}{2})$ aliases from $(FS3 - CW * \frac{1}{2})$ to $\frac{1}{2} * SR$. There should be no aliasing of the COFDM wanted signal into itself. That is, as shown in FIGURE 5, the lower end of the wanted signal, at $(IF1 - SB * \frac{1}{2})$ aliases to $(FS3 + SB * \frac{1}{2})$, and it is therefore desirable that the method adheres to the following condition:

$$(FS3 + SB * \frac{1}{2}) + \Delta < \frac{1}{2} * SR;$$

where Δ again is a possible offset.

However, in this case, the potentially interfering unwanted signal at $(IF1 + N)$ aliases to $(FS3 - N)$, and cannot alias into the wanted signal.

FIGURE 6 illustrates a further alternative to that shown in FIGURE 3. Here, the sampling rate SR has been chosen such that the whole of the tuner pass-band appears within the frequency range from $k * SR$ to $(k + \frac{1}{2}) * SR$, where k is an integer. After sub-sampling, the whole tuner pass-band appears, non-inverted in this case, in the frequency range from 0 to $\frac{1}{2} * SR$, with center frequency $FS4$, where $FS4$ equals $(IF1 - k * SR)$.

As in the example of FIGURE 3, the unwanted NICAM signal remains outside the signal band centered on $FS4$ after this downconversion, and can be filtered out in the demodulator 14.

FIGURE 3 shows the sampling rate SR chosen such that the whole of the tuner pass-band appears within the frequency range from $(k - \frac{1}{2}) * SR$ to $k * SR$. FIGURE 6 shows the sampling rate SR chosen such that the whole of the tuner pass-band appears within the frequency range from $k * SR$ to $(k + \frac{1}{2}) * SR$. FIGURES 4 and 5 show the sampling rate SR chosen such that the tuner pass-band appears largely (but not entirely) within the frequency range from $(k - \frac{1}{2}) * SR$ to $k * SR$. However, it is also possible to choose the sampling rate such that the tuner pass-band appears largely but not entirely within the frequency range from $k * SR$ to $(k + \frac{1}{2}) * SR$, with the same constraints.

To facilitate a better understanding of the present invention, the above-described cases will now be illustrated for the case of a received COFDM signal, which has been downconverted in a first stage to a first intermediate frequency of 36.167 MHz, with a pass-bandwidth of 9.40 MHz. The actual wanted signal bandwidth is 7.61 MHz, centered at the intermediate frequency of 36.167 MHz. The nearest adjacent interference signal is a NICAM signal at $(36.167 + 4.1981) = 40.3651$ MHz.

Choosing a sampling rate, SR, of 20.5 Mega samples per second (Ms/s) means that the whole of the pass band from 31.467 MHz to 40.687 MHz falls within the range from 1.5SR to 2SR, and there is no aliasing of any part of the pass-band into any other. This means that the unwanted signal can be filtered out.

Alternatively, choosing a sampling rate of 21.0 Ms/s means that the lower end of the pass-band falls below 1.5SR, as shown in FIGURE 5. In this case, the lower edge of the pass-band at 31.467 MHz aliases to $(1.5SR - 31.467) = 0.033$ MHz below 0.5SR, while the lower edge of the wanted signal band, at $(36.167 - 7.61 * \frac{1}{2})$ MHz aliases to $(36.167 - 7.61 * \frac{1}{2} - 1.5SR) = 0.862$ MHz below 0.5SR. Thus there is no interference, and the unwanted signal can be filtered out.

Choosing a sample rate of 20 Ms/s means that, as in FIGURE 4, the upper end of the pass-band aliases into the output, and the unwanted signal can potentially interfere with the wanted signal. In this case, the upper edge of the wanted band at $(36.167 + 7.61 * \frac{1}{2}) = 39.972$ MHz aliases to 0.028MHz, while the unwanted signal at $(36.167 + 4.1981) = 40.3651$ MHz aliases to 0.3651 MHz, which is within the wanted band. This will mean that the following condition cannot be met, for any value of Δ :

$$(N-FS2) + \Delta < (FS2-SB * \frac{1}{2}).$$

Summary

A novel method and apparatus for receiving radio frequency signals has been described, wherein the method utilizes an inventive receiver circuit that downconverts an input signal so that interference components of the input signal can be easily removed at baseband. Specifically, the inventive receiver circuit operates by sub-sampling a first intermediate frequency signal in such a way that an unwanted signal is not aliased into a wanted signal, and can therefore be filtered therefrom after sub-sampling. Thus, the present invention allows the use of a relatively simple tuner, with a single downconversion stage, without imposing excessive requirements on the filtering in the tuner. The disclosed methods and apparatus can be utilized with a number of communication systems, including, without limitation, a television communication system.

A number of embodiments have been described. Nevertheless, it will be understood that various modifications may be made without departing from the spirit and scope of the

invention. For example, the present inventive method and apparatus can be implemented in software, hardware, or in a software/hardware combination. Furthermore, the present inventive method and apparatus can be used in virtually any type of communication system. Its use is not limited to a European DVB-T standard-based communication system. Alternatively, the present invention can be used in a North American television standard-based communication system. Accordingly, it is to be understood that the invention is not to be limited by the specific illustrated embodiment, but only by the scope of the appended claims.

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